What is claimed is:

1. A process comprising:

preprocessing audio data to determine parameters associated with time scaling of the audio data;

providing the audio data and the parameters to a device; and

having the device use the parameters in time scaling the audio data to generate time-scaled audio, wherein using the parameters in the time scaling requires less processing power than would time scaling of the audio data without using the parameters.

2. The process of claim 1, wherein the device uses the audio data and the parameters to perform real-time time scaling of the audio data.

3. The process of claim 1, wherein providing the audio data and parameters comprises recording the audio data and the parameters on a storage media that the device can read, and the device accessing the storage media to read the audio data and the parameters.

4. The process of claim 3, wherein the storage media is a disk.

5. The process of claim 1, wherein providing the audio data and the parameters comprises transmitting the audio data and the parameters via a network to the device.

6. The process of claim 1, wherein:

the audio data comprises a plurality of input frames; and

the parameters comprise one or more offsets for each input frame, each offset identifying for an associated input frame a block of samples for use in generating time-scaled data from the associated input frame.

7. The process of claim 6, wherein the parameters comprise a plurality of offsets for each

input frame, the plurality of offsets for each input frame corresponding to different time scales.

8. The process of claim 1, wherein the device performs the preprocessing of the audio data to determine the parameters and stores the audio data and the parameters for later use during real-time time scaling of the audio data.

9. The process of claim 1, wherein:

the audio data comprises a plurality of input frames; and

one or more of the parameters classify respective audio contents of the input frames.

10. The process of claim 9, wherein the parameters identify which of the input frames represent silence.

11. The process of claim 9, wherein having the device use the parameters comprises processing the input frames that the parameters indicate represent silence differently from processing of the input frames that the parameters indicate are not silence.

12. The process of claim 1, wherein a voice mail system performs the preprocessing of the audio data to determine the parameters associated with time scaling of the audio data.

13. The process of claim 12, wherein the device comprises a telephone that receives audio data and the parameters from the voice mail system.

14. The process of claim 1, wherein a server performs the preprocessing of the audio data to determine the parameters associated with time scaling of the audio data.

15. The process of claim 14, wherein the device comprises a telephone that receives audio data and the parameters from the server.

16. The process of claim 1, wherein the device comprises a server that performs the preprocessing of the audio data to determine the parameters associated with time scaling of the audio data, stores the audio data and the parameters for later use, and performs real-time time scaling to provide the time-scaled audio data to a player.

17. A process for time scaling of audio, comprising:

receiving a frame of audio data with parameters indicating a relation between offset and time scale;

using the parameters to determine an offset that corresponds to a selected time scale; and

generating a time-scaled frame using samples that are in a block identified by the offset.

18. The process of claim 17, wherein the parameters comprise a plurality of preprocessed offsets that respectively correspond to a plurality of time scales.

19. The process of claim 18, wherein using the parameters comprises interpolating between the preprocessed offsets to determine the offset corresponding to the selected time scale.

20. The process of claim 17, further comprising a listener selecting the selected time scale for presentation of the audio.

21. An audio data structure, comprising:

a plurality of frames respectively corresponding to sections of audio, each frame comprising a plurality of samples of the corresponding section of audio; and

one or more parameters for each frame, the parameters providing information that reduces an amount of processing power needed for time scaling the audio data.

22. The audio data structure of claim 21, wherein the one or more parameters for a frame identify a block of the samples that is used to generate time-scaled data.

23. The audio data structure of claim 21, wherein each parameter for a frame identifies a block of the samples that is used to generate time-scaled data from the frame.

24. The audio data structure of claim 21, wherein one or more parameters for a frame comprises a plurality of offsets that respectively correspond to a plurality of time scales, each offset identifying a block of the samples that are used to generate time-scaled data that corresponds to the time scale corresponding to the offset.

25. The audio data structure of claim 21, wherein one or more parameters indicate which of the frames correspond to silent sections of the audio.